Patent Docket No.: 977-024(RIC 99-060)

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Inventor: Gallant et al.

March 27, 2002

Serial No: 09/436,796

Group Art Unit: 2662

Filing Date: 11/08/99

Examiner: Joe Logsdon

Title: METHOD AND SYSTEM FOR DYNAMIC GATEWAY SELECTION IN AN IP TELEPHONY NETWORK

RESPONSE

I HEREBY CERTIFY THAT THIS CORRESPONDENCE IS BEING DEPOSITED WITH THE UNITED STATES POSTAL SERVICE AS FIRST CLASS MAIL ADDRESSED TO ASSISTANT COMISSIONER FOR PATENTS, WASHINGTON, D.C. 20231 ON MARCH 27, 2002.

Tammy S. Senecal

Attention: Board of Patent Appeals and Interferences
Assistant Commissioner of Patents and Trademarks
Washington, DC 20231

Sir:

BRIEF ON APPEAL

This Brief supports the appeal to the Board of Patent Appeals and Interferences from the final rejection dated August 27, 2001, in the application listed above. Appellants filed the Notice of Appeal on January 28, 2002, and now submit this Brief in triplicate, as required by 37 C.F.R. § 1.192(a).

I. REAL PARTY IN INTEREST

WorldCom, Inc., as assignee of U.S. Patent Application Serial No. 09/436,796, is the real party in interest.

II. RELATED APPEALS AND INTERFERENCES

There are no related appeals or interferences pertaining to the above-identified application.

III. STATUS OF CLAIMS

A. Claims 1-28 are Finally Rejected

Claims 1-28 have been finally rejected Under 35 U.S.C. §112, first paragraph for lack of adequate written description and for lack of enablement. Claims 1-15 and 19-22 have been finally rejected under 35 U.S.C. §103(a) for obviousness over Schultzrinne, "A Comprehensive Multimedia Control Architecture for the Internet," 1997 IEEE, pages 65-76 ("Schultzrinne") in view of U.S. Patent No. 5,930,348 to Regnier et al. ("Regnier"), and U.S. Patent No. 5,883,894 to Patel et al. ("Patel"). Claims 16-18 have been finally rejected under 35 U.S.C. §103(a) for obviousness over Schultzrinne in view of Regnier. The Appellants note that claims 23-28 were not rejected under 35 U.S.C. § 103(a).

B. Claims 1-28 Are On Appeal

The decision of the U.S. Patent and Trademark Office ("the Patent Office") that finally rejected claims 1-28 is hereby appealed.

IV. STATUS OF AMENDMENTS

No Amendments were filed after the Final Rejection.

V. SUMMARY OF INVENTION

A. Brief Description of the Invention

Appellants' invention is directed to a system and method by which an egress gateway is dynamically selected to establish a communications session over a path supported at least in part by an IP telephony network and a PSTN, PBX or other network. A redirect server (RS) uses a gateway selection method, which includes recording egress gateways which are not available (Specification, at page 4, lines 5-11, and at page 8, lines 17-20). The RS also provides a proxy server with a list of alternate destination gateways ready for selection if the primary choice is unavailable (Specification, at page 5, lines 5 - 6). The RS maintains the list of unavailable destination gateways to avoid the repeated selection of an unavailable gateway

(Specification, at page 4, lines 8 - 11). A gateway is defined as an interface between an IP network and another network (Specification, at page 1, lines 13 - 14).

B. Problems in the Prior Art

For the last century or so, most telephony services have been provided by the public switched telephone network (PSTN). The PSTN is comprised of circuit switched networks that provide users with a dedicated, end-to-end circuit connection for each call. Circuit switched networks are optimized to carry voice traffic (Specification, page 2, lines 14 – 16). However, because of the emergence of the Internet and because of the increasingly computerized business environment, telecommunications customers are demanding integrated voice and data services. For example, the PSTN now carries more IP data traffic than voice traffic (Specification, page 2, line 17 – page 3, line 5). In response to these trends, telecommunications networks that employ packet switching instead of the traditional circuit switched technologies are being deployed. Packet switching is more efficient than circuit switching when it comes to data, because packet switching is session-based rather than connection based. However, because there are a huge number of telephones that are still being serviced by the PSTN, it is imperative that IP telephony networks interface with the PSTN. The interface between IP telephony networks and the PSTN, private PBX networks, and other networks is usually implemented using gateways. Gateways convert PSTN signaling and PSTN traffic into IP telephony formats, and vice-versa (Specification, page 3, lines 12 – 20).

One problem the present invention seeks to address involves the availability, selection, and reliability of egress gateways. The selection of a destination gateway must be done quickly and efficiently or quality of service will suffer. Delays are further exacerbated when a selected gateway is unavailable. Thus, the availability status of egress gateways should be monitored and maintained. (Specification, page 3, line 19 - page 4, line 2).

C. Detailed description of the Present Invention

One aspect of Appellants' invention is directed to a method for routing calls to a destination gateway to establish a communication session between a source user agent and a

destination user agent (Claim 1, Specification, at page 4, lines 12-19, at page 9, lines 4-20, Figure 1, Figure 2, steps 702-724, and Figure 3, steps a – n). The session is established over a path supported at least in part by a telephone network and an IP network (Figure 1 and Specification, at page 5, lines 5-11). The IP network includes a plurality of ingress and destination gateways, at least one proxy server, and at least one redirect server (Specification, at page 9, lines 4-20).

The method includes the step of having a proxy server receive a call setup request from the source user agent (Figure 1, reference numeral 1, Specification, at page 5, lines 1-2, and at page 10, lines 1 – 9). The source user agent is disposed in a public switched telephone network (Specification, at page 10, lines 6 – 9). The call setup request identifies the destination user agent (Specification, at page 10, lines 6 – 9, Table I). The received call setup request is forwarded by the proxy server to a redirect server (Figure 2, Specification, at page 10, lines 10 – 18). The proxy server receives either routing information, or a request failure response, from the redirect server (Figure 2, Specification, at page 10, lines 10 – 20). When the proxy server receives routing information from the redirect server, it proxies the call setup request to a destination gate way selected from the routing information (Figure 2, Specification, at page 11, lines 1 – 2). The selected destination gateway is in communication with a public switched telephone network that includes the destination user agent (Specification, at page 10, lines 6 – 9).

After proxying the call setup request to the selected destination gateway, the proxy server waits for a response from the selected destination gateway. If a response is received from the selected destination gateway within a predetermined time, a communications session is established between the source user agent and the destination user agent using the selected destination gateway (Specification, at page 11, lines 1 – 10). If a response is not received within the predetermined time, the call setup request is sent to a second destination gateway. This gateway is also selected from the routing information (Specification, at page 11, lines 13 – 14). The proxy server reports the failure of the first selected destination gateway to the redirect server (Specification, at page 11, lines 7 – 8). The second destination gateway can

also communicate with a public switched telephone network that includes the destination user agent (Specification, at page 10, lines 15 – 17).

Another aspect of Appellants' invention is directed to a system for allowing a call to be completed in a communication session between a called party and a called party. The system includes a first telephony system that includes at least one service user agent, and a second telephony system that includes at least one destination user agent. An IP network is connected between the first and second telephone systems. A plurality of ingress gateways interfaces the IP network to the first telephone system. A plurality of egress gateways interfaces the IP network to the second telephone system. An IP telephony proxy server selects one the plurality of egress gateways to complete the call. An IP redirect server provides routing information to the IP telephony proxy server. A Network Management System (NMS) receives and stores status changes of destination gateways. The NMS is in communication with the IP redirect server. (Specification, at page 4, lines 15-20, Figure 1).

Another aspect of Appellants' invention is directed to a method for detecting an available destination gateway from a plurality of destination gateways in an IP network (Specification, at page 7, lines 11 – 15, Claim 19). The IP network is used for completing a communications session between a source user agent in one public switched telephone network and a destination user agent in another public switched telephone network (Figure 1). The source user agent provides a call setup request that identifies the destination user agent (Specification, at page 10, lines 6 – 9, Table I). The method includes the step of transmitting a message to one of the plurality of destination gateways from a server to ascertain an availability status of the destination gateway. The method also includes the step of waiting for an acknowledge response from the destination gateway for a predetermined period of time. If the acknowledge response is received within a predetermined period of time, it is determined that the destination gateway is available. If not, a message is transmitted to a succeeding gateway of the plurality of gateways. The succeeding gateway can communicate with the public switched telephone network that includes the destination user agent (Specification, at page 7, lines 3 – 10, at page 24, line 1 – page 25, line 4, and Figure 4).

VI. ISSUES

Issues presented for consideration in this Appeal are:

A. Whether claims 1-28 are properly rejected under 35 U.S.C. § 112, first paragraph.

B. Whether claims 1-15 and 19-22 are properly rejected under 35 U.S.C. § 103 for obviousness where the applied references would not have been properly combinable.

C. Whether claims 1-15 and 19-22 are properly rejected under 35 U.S.C. § 103 for obviousness where the combination of the applied references does not teach the claimed invention.

D. Whether claims 16-18 are properly rejected under 35 U.S.C. § 103 for obviousness where the applied references would not have been properly combinable.

E. Whether claims 16-18 are properly rejected under 35 U.S.C. § 103 for obviousness where the combination of the applied references does not teach the claimed invention.

VII. GROUPING OF CLAIMS

In compliance with 37 C.F.R. § 1.192(c)(5), Applicants state that claims 1 – 28 do not stand or fall together. Claims 1, 16, and 19 are separately patentable. Three groups of claims stand together as patentable. The groups include: (a) claims 1 – 15, and 23 – 26; (b) claims 16 – 18, and 27 – 28; and claims 19 – 22.

VIII. ARGUMENTS

A. Claims 1-28 are patentable under 35 U.S.C. § 112, first paragraph.

1. The Subject Matter Of The Claimed Invention Is Adequately Described In The Specification As Filed.

Claims 1, 16, and 19 recite similar limitations that a call setup request originates from the source user agent (SUA) and identifies the destination user agent (DUA). The Examiner asserts that this claimed limitation lacks an adequate written description under 35 U.S.C. §112, first paragraph. In doing so, the Examiner asserts that the specification fails to describe the SIP INVITE message as modified in accordance with the present invention. The Examiner also asserts that the term "ingress gateways" as recited in the claims, and the manner in which they are used, lacks adequate written description.

To satisfy the written description requirement, a patent specification must describe the claimed invention in sufficient detail that one skilled in the art can reasonably conclude that the inventor had possession of the claimed invention as of the filing date. *Vas-Cath, Inc. v. Mahurkar*, 19 USPQ2d 1111, 1116, 1117 (Fed Cir. 1991). The Applicants can show possession of the claimed invention by describing the claimed invention using descriptive means such as words, structures, figures, diagrams, and formulas that fully set forth the claimed invention. *Lockwood v. American Airlines, Inc.*, 41 USPQ2d 1961, 1966 (Fed. Cir. 1997).

One skilled in the art, upon reading Table 1 and the associated text on page 12, would reasonably conclude that the inventor had possession of the claimed invention as of the filing date. Table 1 lists the required INVITE message parameter fields. The second parameter field in Table 1 is the "To" parameter field. It contains the address of the recipient of the request. The third parameter field in Table 1 is the "From" parameter field. The "From" parameter field contains the address of the initiator of the request. The text on page 12 of the specification immediately below Table 1 states: "The SIP INVITE is addressed to the called party DUA 103 at a proxy address at the SPS 106. The SIP INVITE specifies the real IP

address of the DUA 103." The Appellants are at a loss in trying to understand how they could have described the modified SIP INVITE message any more explicitly.

With respect to the Examiner's assertion that the term ingress gateways, and the manner in which they are used, lacks adequate written description, Appellants first point out that this term was employed in the claims and specification as filed (Specification, at page 1, lines 12 – 13, at page 3, lines 14 – 19, and Claims 1-22). Thus, one skilled in the art would reasonably conclude that the inventor had possession of the claimed term "ingress gateways" as of the filing date. The specification points out that an IP network must interface with origination (e.g., ingress) and destination (egress) switches in the PSTN. The interface is well known in the art as a gateway. A gateway performs code and protocol conversion between two otherwise incompatible networks to provide the necessary link (Specification, at page 3, lines 14 – 19).

For the reasons provided above, the rejection for lack of adequate written description under 35 U.S.C. §112, first paragraph is improper, and should be withdrawn.

2. <u>The Specification Describes How To Make and How To Use The Claimed Invention.</u>

As noted above, the Examiner asserts that claims 1, 16, and 19 lack enablement under 35 U.S.C. §112, first paragraph because the specification fails to describe the SIP INVITE message as modified in accordance with the present invention. According to the Examiner, the term "ingress gateways," and the manner in which they are used, also lacks enablement. Both of these terms were discussed above with respect to the written description requirement.

The test of enablement is "whether one reasonably skilled in the art could make or use the invention from the disclosure in the patent coupled with information known in the art without undue experimentation. <u>A patent need not teach, and preferably omits, what is well known in the art</u>." *In re Buchner*, 18 USPQ2d 1331, 1331 (Fed. Cir. 1991); *Hybritech, Inc. v. Monoclonal Antibodies, Inc.*, 231 USPQ 81, 94 (Fed. Cir. 1986). The amount of guidance

needed to enable an invention is inversely related to the amount of knowledge in the state of the art as well as the predictability in the art. *In re Fisher,* 166 USPQ 18, 24 (CCPA 1970).

The term "ingress gateway" is defined as an origination gateway connecting a IP network to a PSTN, PBX, or other network. A gateway performs code and protocol conversion between two otherwise incompatible networks to provide the necessary link between those networks (Specification, at page 1, lines 12 – 13, and at page 3, lines 14 – 19). Further, the term "gateway" is also enabled because it is a well known term of art that anyone of ordinary skill in the art recognizes and understands. As noted above, a patent need not teach, and preferably omits, what is well known in the art. A gateway is a standard element in the Internet. The Internet has been around since the 1960's. Because gateways have been extensively used, those of ordinary skill in the art need virtually no guidance to make or use an ingress gateway. Thus, it is both shocking and mystifying to have the Patent Office maintain that the term "gateway" is not enabled.

Claims 1, 16, and 19 recite that a call setup request originates from the source user agent (SUA) and identifies the destination user agent (DUA). The Examiner also asserts that this claimed limitation is not enabled under 35 U.S.C. §112, first paragraph. As noted above, Table 1 lists the INVITE required parameter fields. The second parameter field in Table 1 is the "To" parameter field. It contains the address of the recipient of the request. The third parameter field in Table 1 is the "From" parameter field. The From parameter field contains the address of the initiator of the request. The text on page 12 of the specification immediately below Table 1 states: "The SIP INVITE is addressed to the called party DUA 103 at a proxy address at the SPS 106. The SIP INVITE specifies the real IP address of the DUA 103." There is no experimentation involved whatsoever in sending and receiving a "call setup request that identifies the destination user agent" because Table I explicitly discloses the message format of the INVITE request. Given the guidance provided on page 12 and in Table 1, any entry level computer programmer could modify the standard SIP INVITE message in accordance with the claimed invention.

For the reasons provided above, the rejection for lack of enablement under 35 U.S.C. §112, first paragraph is improper, and should be withdrawn.

B. Description of the prior art cited by the Examiner.

1. Schultzrinne.

Schultzrinne is related to a control architecture for the Internet. The architecture employs two independent but related protocols: the Session Initiation Protocol (SIP), which is used to invite participants to multimedia sessions; and the Real-Time Stream Protocol (RTSP) which is used to control playback and recording for stored continuous media. On page 68, Schultzrinne describes the method for establishing a call over the Internet. The method includes the steps of locating the called terminal, agreeing on the media and the encoding to be used during the call, and determining if the called party desires to be reached. The step of locating the called party includes the step of determining the name of the called party. This method is shown in Figure 1. In this process, the client attempts to contact a SIP server. If that fails, it tries to connect to a SMTP server. If that fails, the call invitation is sent via an e-mail message. If the called party is not at the SIP server named, the SIP server may initiate a redirection response. On page 69, Schultzrinne describes what is traditionally known as a "find-me" feature, wherein the SIP server searches for a party by multicasting all of the addresses corresponding to the party. The teaching of Schultzrinne is not directed to the problem of selecting an available gateway.

2. Regnier.

Regnier is related to call routing control in circuit switched networks. As shown in Figure 1, the circuit switched network employs Signaling System No. 7 (SS7), which is commonly used by most PSTNs. SS7 is used to control service switching points (SSP) that interconnect the trunks (L1-L10) in circuit switched network 10. Each SSP is connected to a service control point/network (SCP/NP) processor (See Figure 2, Figure 5, and Figure 6). A SCP/NP is the network resource used to implement the dynamic routing method described by

Regnier. In the example provided, a first SSP attempts to establish a call with a neighboring SSP. Unfortunately, the call is unsuccessful because the neighboring SSP has exhausted all of its circuits. The first SSP notifies the SCP/NP of the unsuccessful attempt. The SCP/NP identifies the unavailable trunk, updates its database, and finds an alternative route. The SCP/NP transmits a message to the first SSP identifying the alternative route (Regnier, at column 8, line 50 – column 9, line 62).

Regnier also discloses a situation wherein facilities in an adjacent "other" network are not available. The "other network" node 26 is operated by a different regional operating company. From the context it is clear that the other network is also a circuit switched network, and not an IP network. SSPs may also be employed as interfaces between two circuit switched networks (Regnier, at column 11, lines 11 – 38). However, SSPs are not gateways as defined by the present invention because SSP elements are components of Signaling System No. 7 (SS7) that may couple adjacent circuit switched networks. They are not configured to provide an interface connecting an Internet Protocol (IP) network to a PSTN, PBX, or some other network as defined in the present invention.

3. Patel.

Patel is directed to auto-negotiation logic used in an interconnection device used to connect different terminals in a LAN. Typically, LAN terminals are designed to support a specific LAN technology. For example, the Ethernet network standard supports various technologies such as the 10 Base-T standard (10 Mb/s CSMA/CD over twisted pair telephone wire), the 10 Base-F standard (10 Mb/s CSMA/CD over optical fiber), the 100 Base-TX standard (100 Mb/s CSMA/CD over two pairs of shielded twisted pairs), and a number of other technologies. The auto-negotiation function is deployed on a shared intermediate LAN device that interconnects a plurality of terminals employing the various technologies described above. The shared unit connects ports to the shared unit in a round-robin fashion using the auto-negotiation logic. It is not at all clear why this reference is relevant.

C. <u>Claims 1-15 and 19-22 are patentable under 35 U.S.C. § 103(a) because Schultzrinne, Regnier and Patel would not have been properly combinable.</u>

The PTO may not properly combine prior art references in order to establish *prima facie* obviousness unless there is "some suggestion for doing so found either in the references themselves or in the knowledge generally available to one of ordinary skill in the art." *In re Jones,* 21 USPQ2d 1941, 1943 – 44 (Fed. Cir. 1992); See also *In re Geiger,* 2 USPQ2d 1276, 1278 (Fed. Cir. 1987). Thus, obviousness cannot be demonstrated by combining prior art references absent some teaching, suggestion or incentive supporting the combination. Also, there can be no suggestion or motivation to make a proposed modification if the proposed modification renders the prior art unsatisfactory for its intended purpose. *In re Gordon,* 221 USPQ 1125 (Fed. Cir. 1984). It has also been established that the teachings of prior art references are not sufficient to render the claims prima facie obvious if a proposed modification of the prior art would change the principle of operation of the prior art being modified. *In re Ratti,* 123 USPQ 349 (CCPA 1959). It is also well established that the Examiner must avoid impermissible hindsight. The Examiner must forget about what he was taught about the claimed invention and cast his mind back to the time the invention was made, to occupy the mind of one skilled in the art who is guided only by the references and the then-accepted wisdom in the art. *W.L. Gore & Associates, Inc. v Garlock, Inc.,* 220 USPQ 303 (Fed. Cir. 1983), *cert. denied,* 469 U.S. 851 (1984).

The Examiner states that "it would have been obvious to one of ordinary skill in the art to modify the teaching of Schultzrinne so that the destination server is a gateway, as in Regnier et al., and...because use of gateways allows the systems of caller and callee to use different protocols, and restricting the wait time avoids situations in which the proxy server must wait indefinitely because of a link failure." (See Paper 8, Final Office Action, page 6, lines 15 – 21). The rationale provided by the Examiner for combining the references is, by and large, incomprehensible. The Examiner does not point out in the references, or in the knowledge generally available to one of ordinary skill in the art, where his stated motivation to combine can be found. The statement appears to be poorly thought out, and completely off-the-cuff.

Appellants respectfully submit that the Examiner has, with twenty-twenty hindsight, merely restated the advantages of Appellants' claimed invention. The Appellants point out that the concept of using gateways to allow the systems of caller and callee to use different protocols is taught by the present invention (Specification, at page 3, lines 15 – 19). The Appellants also point out that the concept of restricting wait time is also taught by the present invention (Specification, at page 7, lines 11 – 15). Thus, the Examiner relies on the combination of Schultzrinne, Regnier, and Patel through impermissible hindsight.

There is no suggestion or motivation to make the proposed modification because the proposed modification renders the prior art unsatisfactory for its intended purpose. Schultzrinne teaches an Internet control architecture. Everyone of ordinary skill in the art knows that the Internet is a packet switched network. Regnier teaches a method for routing calls in a circuit switched network. The combination of Schultzrinne and Regnier renders both of these references unsatisfactory for their intended purpose. It is impossible to employ a circuit switched routing control scheme in a packet switched network because the use of circuit switched routing will not work in a packet switched network. As noted above, circuit switched networks establish end-to-end circuits. Packet switched networks do not. Individual packets in a given session may traverse completely different paths between end users. Trying to employ Regnier's SS7 call routing in Schultzrinne's Internet control architecture is absolutely meaningless.

The teachings of prior art references are not sufficient to render the claims prima facie obvious because the proposed modification of the prior art changes the principle of operation of the prior art being modified. As noted above, circuit switched networks and packet switched networks operate in a completely different manner. It is impossible to employ a routing control scheme for a circuit switched network in a packet switched network without changing Schultzrinne's principle of operation. The Examiner does not explain why, or how for that matter, one of ordinary skill in the art would use circuit switched routing in conjunction with SIP messaging to establish and maintain packet switched sessions over the Internet. The proposed combination is ridiculous.

The Appellants find it extremely difficult to comment on the Examiner's use of the Patel reference because it has no apparent relevance to either Schultzrinne, Regnier, or the claimed invention. As described in detail above, the Patel reference discloses an auto-negotiation state machine that is employed in a LAN interconnection device. The proposition that one of ordinary skill in the art would look to combine Patel's auto-negotiation state machine with either Schultzrinne or Regnier strains credulity.

For the reasons provided above, the rejection of claims 1 – 15 and 19 – 22 as being unpatentable for obviousness under 35 U.S.C. §103(a) is improper, and should be withdrawn.

D. Even if Schultzrinne, Regnier and Patel were combinable, which they were not, the combination of these references does not teach the claimed invention.

For all the reasons stated *supra*, there would have been no reason to combine these references. However, even if such a combination were proper, it still would not teach the present invention as recited in claim 1 and claim 19.

As noted above, claim 1 is directed to a method for routing calls to a destination gateway to establish a communication session call in a telecommunications network between a source user agent and a destination user agent over a path supported at least in part by a circuit switched telephone network and an IP network. The IP network includes a plurality of ingress and destination gateways, at least one proxy server, and at least one redirect server. Claim 1 also includes the step of proxying a call setup request by a proxy server to a destination gateway selected from routing information received from a redirect server. The selected destination gateway is configured to communicate with a public switched network that includes the destination user agent. Neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest selecting a destination gateway from a plurality of destination gateways. As noted above, while Schultzrinne includes proxy servers and redirect servers, but these are not configured to select a destination gateway from a plurality of destination gateways as claimed. Regnier has nothing whatsoever to do with IP

networks. Thus, neither of these references have any teaching whatsoever about proxying call setup requests to a destination gateway selected from routing information received from a redirect server. Patel is completely irrelevant. Furthermore, neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest the step of waiting for a response from the selected destination gateway upon proxying the call setup request, as recited in claim 1 and defined in the specification.

Claim 1 includes the step of establishing a communication session using the selected destination gateway if the selected destination gateway responds within a predetermined period of time. Neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest this limitation either. Finally, neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest a method that includes the step of sending the call setup request to a succeeding destination gateway selected from routing information and reporting failure of the selected destination gateway to the redirect server. As recited in claim 1, the succeeding destination gateway can communicate with a public switched telephone network that includes the destination user agent.

Claim 19 is directed to a method for detecting an available destination gateway from a plurality of destination gateways in an IP network in order to complete a communications session between a source user agent in one public switched telephone network, and a destination user agent in another public switched telephone network. Neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest this method. Claim 19 includes the step of transmitting a message from a server to one of the plurality of destination gateways to ascertain an availability status of that particular destination gateway. Neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest the above limitations.

Claim 19 includes the step of waiting for an acknowledge response from the one of the plurality of destination gateway for a predetermined period of time. Claim 19 includes the step of determining if one of the plurality of gateways is available if the acknowledge

response is received within the predetermined period of time. Claim 19 also includes the step of transmitting a message to a succeeding gateway of the plurality of gateways. The succeeding gateway is configured to communicate with the public switched telephone network that includes the destination user agent. Neither Schultzrinne, Regnier, nor Patel, whether taken individually or in combination, teach or suggest the above limitations.

The Appellants are also at a loss in trying to understand the Examiner's rejection of the dependent claims. For example, claim 2 recites the limitation of repeating steps (d) to (g) until a destination gateway is determined to be available for establishing the communication session or until all destination gateways from said routing information have been determined to be unavailable. The Examiner does not point out where in the references this step can be found. Claim 3 also depends from claim 1 and further includes the step of recording a destination gateway status as out-of-service if the response from the destination gateway is not received within the predetermined time. The Examiner also does not point out where in the references this step can be found. As a matter of fact, the Appellants could easily make this point about every dependent claim. The Examiner does make separate arguments about claim 6 and claim 8, but again, he does not say which reference supplies the missing claim limitations. In each case, the Examiner merely asserts that it would be obvious for reasons either invented out of thin air, or for reasons based on impermissible hindsight. The Appellants find the arbitrary nature of the Examiner's rejection most disturbing.

For the reasons provided above, the rejection of claims 1 – 15 and 19 – 22 as being unpatentable for obviousness under 35 U.S.C. §103(a) is improper, and should be withdrawn.

E. <u>Claims 16-18 are patentable under 35 U.S.C. § 103(a) because Schultzrinne and Regnier would not have been properly combinable.</u>

The Examiner's rationale for combining the cited references does not point out where in the references, or where in the knowledge generally available to one of ordinary skill in the art, his stated motivation is found (See the Final Office Action, Paper 8, at page 9). Again, the reasons provided by the Examiner for combining the references are poorly thought out,

and poorly communicated. The Examiner does not point out in the references, or in the knowledge generally available to one of ordinary skill in the art, where his stated motivation to combine can be found. Once again the Examiner states that "it would have been obvious to one of ordinary skill in the art to modify the teaching of Schultzrinne so that the destination server is a gateway..." As noted above, the Appellants point out that the concept of using gateways to allow the systems of caller and callee to use different protocols is taught by the present invention (Specification, at page 3, lines 15 – 19). Thus, the Examiner relied on the combination of Schultzrinne and Regnier through impermissible hindsight.

As noted above with respect to claim 1 and claim 19, the Examiner's proposed combination of Schultzrinne and Regnier not only renders the cited prior art unsatisfactory for its intended purpose, but it also changes the principles of operation for each of the cited references.

For the reasons provided above, the rejection of claims 16 – 18 as being unpatentable for obviousness under 35 U.S.C. §103(a) is improper, and should be withdrawn.

F. Even if Schultzrinne and Regnier were combinable, which they were not, the combination of these references does not teach the claimed invention.

For all the reasons stated *supra*, there would have been no reason to combine these references. However, even if such a combination were proper, it still would not teach the present invention as recited in claim 16. Claim 16 is directed to a system that includes an IP network connected between a first telephony system and a second telephony system. The system further includes a plurality of ingress gateways for interfacing the IP network to the first telephony network, and a plurality of egress gateways for interfacing the IP network to the second telephony network. The system also includes an IP telephony proxy server for selecting of one of the plurality of egress gateways for completing the call based on routing information received by the IP telephony proxy server. As recited in claim 16, the IP telephony proxy server makes the selection after receiving a call setup request from the source user agent. The call setup request identifies the destination user agent. None of these

limitations can be found in the cited references. Neither Schultzrinne nor Regnier, whether taken individually or in combination, teach or suggest these limitations.

Neither Schultzrinne nor Regnier, whether taken individually or in combination, teach or suggest a system that includes a Network Management System (NMS) that is in communication with an IP redirection server, as recited in claim 16. The IP redirection server receives and stores status changes of destination gateways. While Schultzrinne does disclose an IP network having redirect servers, he does not disclose redirect servers that are configured to maintain the status of destination gateways. Regnier does not employ redirect servers in any fashion whatsoever. Thus, Regnier does not supply the claimed features that are missing from Schultzrinne.

For the reasons provided above, the rejection of claims 16 – 18 as being unpatentable for obviousness under 35 U.S.C. §103(a) is improper, and should be withdrawn.

IX. CONCLUSION

In conclusion, Applicants request a reversal of each of the grounds of rejection maintained by the Examiner. If there are any other fees due in connection with the filing of this Brief on Appeal, please charge the fees to our Deposit Account No. 50-0289. If a fee is required for an extension of time under 37 C.F.R. § 1.136 not accounted for above, such an extension is requested and the fee should also be charged to our Deposit Account.

Respectfully submitted,

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APPENDIX TO BRIEF ON APPEAL

The claims on appeal are as follows:

1. (Amended) A method for routing calls to a destination gateway to establish a communication session call in a telecommunications network between a source user agent and a destination user agent over a path supported at least in part by a telephone network and an IP network, said IP network including a plurality of ingress and destination gateways, at least one proxy server, and at least one redirect server (RS), said method comprising the steps of:

a) receiving a call setup request at the at least one proxy server from the source user agent, wherein the source user agent is included in a public switched telephone network and the call set up request identifies the destination user agent;

b) forwarding the received call setup request to the redirect server;

c) receiving routing information or a request failure response from the redirect server;

d) proxying the call setup request by the at least one proxy server to a destination gateway selected from said routing information upon receiving the routing information from the redirect server, wherein the selected destination gateway can communicate with a public switched telephone network that includes the destination user agent;

e) upon proxying the call setup request to the selected destination gateway, waiting for a response from the selected destination gateway;

f) upon receiving the response from the selected destination gateway within a predetermined time, establishing a communication session using said selected destination gateway; and

g) if the response is not received within the predetermined time, sending the call setup request to a succeeding destination gateway selected from the routing information and reporting failure of the selected destination gateway to the redirect server, wherein the succeeding destination gateway can communicate with a public switched telephone network that includes the destination user agent.

2. The method as claimed in claim 1, further comprising repeating steps (d) to (g) until a destination gateway is determined to be available for establishing said communication session or until all destination gateways from said routing information have been determined to be unavailable.

3. The method as claimed in claim 1, further comprising the step of recording a destination gateway status as out-of-service if the response from said destination gateway is not received within said predetermined time.

4. The method as claimed in claim 3, wherein said step of recording records said destination gateway status as out-of-service in a gateway information table stored within the RS.

5. The method as claimed in claim 1, wherein said step of receiving a call setup request at the at least one proxy server from the source user agent includes the step of addressing said call setup request to a proxy address of the at least one proxy server.

6. The method as claimed in claim 1, wherein said step of receiving a call setup request at the at least one proxy server from the source user agent includes the step of counting a number of received requests subsequent to said call setup request at the at least one proxy server.

7. The method as claimed in claim 1, wherein the at least one proxy server comprises a Session Initiation Protocol (SIP) proxy server.

8. The method as claimed in claim 1, wherein the at least one proxy server comprises an H.323 gatekeeper.

9. The method as claimed in claim 1, wherein said step of responding to the forwarded call setup request from said at least one proxy server received at the RS includes determining the status of a group of destination gateways.

10. The method as claimed in claim 9, wherein the status of each of said group or destination gateway is one of in-service and out-of-service.

11. The method as claimed in claim 10, wherein if the destination gateway status if recorded as out-of-service in a gateway information table and its associated time value is greater than a current absolute RS time, the gateway address is not added to a routing list of said routing information.

12. The method as claimed in claim 10, wherein if the destination gateway status is recorded as out-of-service in a gateway information table and its associated time value is less than or equal to the current absolute RS time, the gateway address is added to a routing list of said routing information and recorded as in-service.

13. The method as claimed in claim 10, further including the step of sending a message from the at least one proxy server to a network manager to record the status of a destination gateway.

14. The method as claimed in claim 1, further comprising the step of forwarding a request failure response to the source user agent upon receiving the request failure response from the at least one proxy server, and terminating the communication session.

15. The method as claimed in claim 1, further comprising the step of resending the call setup request to the selected destination gateway a predetermined number of times when the response is not received within the predetermined time.

16. A system for allowing a call to be completed in a communication session between a calling party and a called party, which comprises:

a first telephony system including at least one source user agent (SUA);

a second telephony system including at least one destination user agent (DUA);

an IP network connected between said first and second telephony systems;

a plurality of ingress gateways for interfacing said IP network to said first telephony system;

a plurality of egress gateways for interfacing said IP network to said second telephony system;

an IP telephony proxy server for selecting one of said plurality of egress gateways for completing said call based on routing information received by the IP telephony proxy server, wherein the IP telephony proxy server receives a call setup request from the source user agent that identifies the destination user agent;

an IP redirect server for providing the routing information to said IP telephony proxy server; and

a network management system for receiving and storing status changes of destination gateways, said network management system being in communication with said IP telephony proxy server.

17. The system as claimed in claim 16, wherein the IP telephony proxy server is a Session Initiation Protocol (SIP) proxy server.

18. The system as claimed in claim 16, wherein the IP telephony proxy server is an H.323 gatekeeper.

19. A method for detecting an available destination gateway from a plurality of destination gateways in an IP network for completing a communication session between a source user agent in a public switched telephone network and a destination user agent in a public switched telephone network, wherein the source user agent provides a call setup request that identifies the destination user agent, said method comprising the steps of:

a) transmitting a message to one of said plurality of destination gateways from a server to ascertain an availability status of said one of said plurality of destination gateways, wherein said one of said plurality of destination gateways can communicate with the public switched telephone network that includes the destination user agent;

b) waiting for an acknowledge response from said one of said plurality of destination gateways for a predetermined period of time;

c) determining if said one of said plurality of destination gateways is available if said acknowledge response is received within said predetermined period of time; and

d) transmitting said message to a succeeding gateway of said plurality of destination gateways, wherein said succeeding gateway can communicate with the public switched telephone network that includes the destination user agent.

20. The method as claimed in claim 19, further comprising repeating steps (b) to (d) until the availability status of each of said plurality of destination gateways has been determined.

21. The method according to claim 19, wherein if said acknowledge response is not received within a predetermined period of time, said availability status of said destination gateway is said to be out-of-service.

22. The method according to claim 19, wherein if said one of said plurality of destination gateways is determined to be available, then said availability status is determined to be in-service.

23. The method according to claim 1, wherein the routing information identifies at least one destination gateway that can handle the call according to status information tracked by the redirect server.

24. The method according to claim 1, wherein the call setup request identifies the destination user agent by specifying the address of the destination user agent.

25. The method according to claim 24, wherein the address of the destination user agent includes the real IP address of the destination user agent.

26. The method according to claim 1, wherein the redirect server tracks status of at least one destination gateway.

27. The method according to claim 16, wherein the call setup request identifies the destination user agent by specifying the address of the destination user agent.

28. The method according to claim 27, wherein the address of the destination user agent includes the real IP address of the destination user agent.